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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/725,294	MARUMOTO ET AL.	
	Examiner	Art Unit	
	MICHAEL C. COLUCCI	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-3,5,7-11 and 13-17* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-3,5,7-11 and 13-17* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *01 December 2003* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments filed 11/21/2008 have been fully considered but they are not persuasive.

Argument 1 (pages 8-10):

- “received speech to be output by a speaker is not received at a microphone of the speech communication apparatus”

Response to argument 1:

Re claims 8 and 14, Examiner believes that based on the claim language and its reasonable meaning, this limitation to be construed as the concept of using two cellular/mobile phones to communicate with one another. For example, user 1 speaks into the microphone of communication device 1, wherein communication device 1 filters noise and transmits a speech signal. User 2 receives the speech signal from a speaker located on communication device 2, thus rendering received speech to be output by a speaker (on communication device 2) is not received at a microphone of the speech communication apparatus (it is instead originally received at the communication device 1 from user 1). Another words, the microphone that picked up the voice of user 1 is output from the speaker of device 2, as is both well known and demonstrated by Urbanski. The teaching of Urbanski is directly affiliate with noise suppression in a cellular environment and directly applicable to the communication between two mobile devices.

Therefore, Examiner believes that Urbanski in fact clearly appears to teach the use of multiple devices communicating with one another, wherein Urbanski teaches a microphone 101 is coupled to receive an input signal 113, in acoustic form, to produce the input signal in electric form at line 115. The A/D converter 103, operatively coupled to the microphone 101, converts the input signal in electric form at line 115 to a digital signal at line 116. The noise suppression system 105, operatively coupled to the A/D converter 103, produces a noise suppressed signal at line 117 responsive to the input signal at line 116. The audio signal processor 107, operatively coupled to the noise suppression system 105, produces a processed signal at line 119 responsive to the noise suppressed signal at line 117. <u>The transmitter 109, operatively coupled to the audio processor 107, transmits the processed signal at line 119 to produce a transmitted signal in electric form at line 121. The antenna 111, operatively coupled to the transmitter 109, converts the transmitted signal in electric form at line 121 to a transmitted signal in electromagnetic form</u> as represented by reference number 123. The <u>communication unit 100 is preferably a cellular radiotelephone</u>. For example, the <u>cellular radiotelephone may be a digital cellular radiotelephone, such as used in the North American Digital Cellular (NADC) System; the Japan Digital Cellular (JDC) System; or the Group Special Mobile (GSM) System</u>. Alternatively, the <u>communication unit 100 may be a two-way radio, cordless radiotelephone</u>, or <u>a wireless microphone</u> (Col. 3 line 65 – Col. 4 line 4).

Further, Urbanski alone teaches that which is well known in the art, wherein Urbanski teaches acoustic noise suppression in a speech communication system generally serves the purpose of improving the overall quality of the desired audio signal by filtering environmental background noise from the desired speech signal. This speech enhancement process is particularly necessary in environments having abnormally high levels of ambient background noise, such as an aircraft, a moving vehicle, or a noisy factory. One noise suppression technique is a spectral subtraction--or a spectral gain modification--technique. Using this approach, the audio input signal is divided into individual spectral bands by a bank of bandpass filters, and particular spectral bands are attenuated according to their noise energy content. A spectral subtraction noise suppression prefilter utilizes an estimate of the background noise power spectral density to generate a signal-to-noise ratio (SNR) of the speech in each channel, which, in turn, is used to compute a gain factor for each individual channel. The gain factor is used as the attenuation for that particular spectral band. The channels are then attenuated and recombined to produce the noise-suppressed output waveform. In specialized applications involving relatively high background noise environments, most noise suppression techniques exhibit significant performance limitations. One example of such an application is the vehicle speakerphone option to a cellular mobile radio telephone system, which provides hands-free operation for the automobile driver. The mobile hands-free

microphone is typically located at a greater distance from the user, such as being mounted overhead on the visor. The more distant microphone delivers a much poorer signal-to-noise ratio to the land-end party due to road and wind noise conditions. Although the received speech at the land-end is usually intelligible, continuous exposure to such background noise levels often increases listener fatigue. In rapidly-changing high noise environments, a severe low frequency noise flutter develops in the output speech signal which resembles a distant "jet engine roar" sound. This noise flutter is inherent in a spectral subtraction noise suppression system, since the individual channel gain parameters are continuously being updated in response to the changing background noise environment (Urbanski Col. 1 lines 20-63).

2. Applicants arguments with respect to claims 1 and 10 have been considered but are moot in view of the new grounds of rejection. Suzuki US 4420655 A (hereinafter Suzuki) and Todter et al US 5937070 A (hereinafter Todter) have been incorporated to address the amended claims 1 and 10.

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 1-3, 5, 7-9, 13, and 16 are rejected under 35 U.S.C. 103(a) as being unpatentable over Urbanski, US 5,544,250 A (hereinafter Urbanski) in view of Soli et al. US 6563931 B1 (hereinafter Soli) and further in view of Suzuki US 4420655 A (hereinafter Suzuki).

Re claim 1, Urbanski teaches a speech communication apparatus for bi-directional speech communications, comprising:

a speaker (Col. 1 lines 42-56);

a microphone (Fig. 1 item 101);

transmission means for transmitting speech to be transmitted which has been extracted by the transmission-speech signal generation filter (Col. 3 line 65 – Col. 4 line 4).

received-speech clarifying means for adjusting a gain (Fig. 2 item 207) for a received-speech signal to be output by the speaker based on the level of the background sound from the output signal measured by the background sound level measurement means (Col. 1 lines 21-29);

wherein the speech communication apparatus does not comprise more than one microphone (Fig. 1 item 101).

wherein the received speech signal to be output by the speaker is not received at the microphone of the speech communication apparatus (Col. 3 line 65 – Col. 4 line 4).

However, Urbanski fails to teach two signals together, wherein one signal is the received speech and the other signal is the background sound level.

Soli teaches a single microphone technique that is used to provide a directionality to the microphone so that the wearer (user) can optimize the wanted part of the signal, the speech, while decreasing any unwanted part of the signal, the noise, which is not directionally coincident with the speech signal (Soli Col. 2 lines 16-20 & Fig. 1 items 6, 8, and 11).

Additionally, Soli teaches a single microphone 11 providing the noise reference signal and primary input signal, the adaptive filter 16 will tend to cancel desired signal as well as noise if desired signal is present in the input signal while filter 16 is adapting (Soli Col. 7 lines 45-56). The present invention thus allows that a human, such as the user, actuate the adapting mode of the filter 10 when noise alone is present at the microphone, to the best extent possible. For example, the user could wait for a pause in a conversation, or request a pause in a conversation, and actuate the adapting mode during this pause. This allows filter 10 to adapt to characteristics minimizing the noise passing through the filter without causing loss of the desired signal

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Urbanski to incorporate two signals, wherein one signal is the received speech and the other signal is the background sound level as taught by Soli to allow for a system that adapts in order to minimize noise and preserve the quality of a received signal, wherein a user can intervene to enable the

adaptation to maximize the received signal (i.e. hearing aids, phone, etc.) (Soli Col. 7 lines 45-56).

However, Urbanski in view of Soli fails to teach a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal;

a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter

background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone

Suzuki teaches a circuit for compensating for frequency characteristic of microphone output which is arranged to have a combination of a microphone member of the pressure gradient type having a proximity effect represented by a rise in its low frequency range sensitivity as the microphone member approaches closer to a source of sound, and another microphone member of the pressure type developing no proximity effect, to cancel out the occurrence of proximity effect. A change in sensitivity of the pressure gradient type microphone in the low frequency range due to the proximity effect is determined from the level difference between the outputs of the two microphones. A signal representative of the difference is subtracted from the output of

the pressure gradient type microphone thereby to effect a compensation for the proximity effect in the output of the pressure gradient type microphone (Suzuki Abstract & fig. 3).

Further, Suzuki teaches well known uses of a filter used to reduce the proximity effect by applying a frequency scale based on the proximity effect onto a signal (Suzuki Fig. 2,4, and 6), wherein the difference between the output of a microphone and the output of a filter are generated with respect to the proximity effect (Suzuki Fig. 1).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Urbanski in view of Soli to incorporate a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal, a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter, and background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone as taught by Suzuki to allow for noise reduction within a greater frequency range, wherein frequency response is independent of the minimization of the proximity effect, and can therefore reduce noise within the whole frequency range (Suzuki Col. 3

lines 38-52 & Fig. 3) whereby the proximity effect is compensated for through a subtractive filter means (Suzuki Fig. 1).

Re claims 2 and 16, Urbanski teaches the speech communication apparatus of claim 1, further comprising:

received-speech-level measurement means for measuring a level of the received-speech signal at each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3),

wherein the background sound level measurement means measures the level of the background sound in each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3) and the received-speech clarifying means performs loudness compensation in which the gain for the received-speech signal is adjusted (Fig. 2 item 207) in each predetermined frequency band (Col. 1 lines 21-41 & Fig. 3) such that received speech output by the speaker is heard at almost the same intensity in the human auditory sense irrespective of the level of the background sound (Col. 1 lines 21-41 & Fig. 3), and the resultant signal is output to the speaker as the received speech (Col. 1 lines 42-56).

Re claims 3, 7, 13, and 17, Urbanski teaches the speech communication apparatus of claim 1, wherein the speech communication apparatus is a portable, mobile telephone for performing the speech communications by radio communication (Col. 1 lines 42-56).

Re claim 5, Urbanski teaches the speech communication apparatus of claim 4, wherein the microphone is a unidirectional or bi-directional microphone (Fig. 1 item 101).

NOTE: The term microphone disclosed in general, is encompassing of any microphone polar pattern including but not limited to unidirectional or bidirectional microphone. Therefore, it would be necessary to utilize a microphone with a particular polar pattern as a matter of choice if and when directional preference is desirable. Urbanski teaches a general microphone such as a unidirectional or bidirectional microphone on a mobile cellular phone (Col. 1 lines 42-56), where a bidirectional microphone could consist of two unidirectional microphones facing opposite one another.

Re claim 8, Urbanski teaches a speech communication apparatus comprising:

a speaker (Col. 1 lines 42-56);

a microphone (Fig. 1 item 101);

a background sound microphone (Col. 1 lines 21-29);

a background sound level calculator operable to calculate a level of a signal outputted from the adder and a level of the background sound (Col. 1 lines 21-29);

a background sound level filter operable to minimize proximity effect; and a received speech (Fig. 2 item 207 clarifying filter operable to adjust a gain for received speech to be output by the speaker based on the background sound level, wherein the

received speech to be output by the speaker is not received at a microphone of the speech communication apparatus (Col. 1 lines 29-41)

However, Urbanski fails to teach a transmission speech filter operable to reduce a level of a lower frequency component of an output signal from the microphone (Soli Col. 2 lines 1-15);

an adaptive filter operable to estimate speech signals from the background sound microphone (Soli Fig. 1 item 16);

an adder operable to subtract the estimated speech signal from the output of the background sound microphone (Soli Fig. 1 item 17);

two signals, wherein one signal is the received speech and the other signal is the background sound level.

Soli teaches a single microphone technique that is used to provide a directionality to the microphone so that the wearer (user) can optimize the wanted part of the signal, the speech, while decreasing any unwanted part of the signal, the noise, which is not directionally coincident with the speech signal (Soli Col. 2 lines 16-20 & Fig. 1 items 6, 8, and 11).

Additionally, Soli teaches a single microphone 11 providing the noise reference signal and primary input signal, the adaptive filter 16 will tend to cancel desired signal as well as noise if desired signal is present in the input signal while filter 16 is adapting (Soli Col. 7 lines 45-56). The present invention thus allows that a human, such as the user, actuate the adapting mode of the filter 10 when noise alone is present at the microphone, to the best extent possible. For example, the user could wait for a pause

in a conversation, or request a pause in a conversation, and actuate the adapting mode during this pause. This allows filter 10 to adapt to characteristics minimizing the noise passing through the filter without causing loss of the desired signal

However, Urbanski in view of Soli fails to teach the minimization of the proximity effect.

Suzuki teaches a circuit for compensating for frequency characteristic of microphone output which is arranged to have a combination of a microphone member of the pressure gradient type having a proximity effect represented by a rise in its low frequency range sensitivity as the microphone member approaches closer to a source of sound, and another microphone member of the pressure type developing no proximity effect, to cancel out the occurrence of proximity effect (Suzuki Abstract & fig. 3).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Urbanski to incorporate two signals, wherein one signal is the received speech and the other signal is the background sound level as taught by Soli to allow for a system that adapts in order to minimize noise and preserve the quality of a received signal, wherein a user can intervene to enable the adaptation to maximize the received signal (i.e. hearing aids, phone, etc.) (Soli Col. 7 lines 45-56). Additionally, it would also have been obvious to one of ordinary skill in the art at the time of the invention to incorporate the minimization of the proximity effect as taught by Urbanski in view of Soli to allow for noise reduction within a greater frequency

range, wherein frequency response is independent of the minimization of the proximity effect, and can therefore reduce noise within the whole frequency range (Suzuki Col. 3 lines 38-52 & Fig. 3).

Re claim 9, Urbanski teaches the speech communication apparatus of claim 8, further comprising:

transmission means for transmitting an output of the transmission-speech filter as a transmission-speech signal from the speech communications apparatus (Col. 1 lines 13-20).

5. Claims 10 and 11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Urbanski, US 5,544,250 A (hereinafter Urbanski) in view of Suzuki US 4420655 A (hereinafter Suzuki) and further in view of Todter et al US 5937070 A (hereinafter Todter).

Re claim 10, Urbanski teaches a speech communication apparatus for bi-directional speech communications, provided with a handset having at a front face a speaker for outputting received speech and a transmission-speech microphone for collecting speech to be transmitted (Col. 1 lines 42-56), the speech communication apparatus comprising:

transmission means for transmitting speech to be transmitted which has been extracted by the transmission-speech signal generation filter (Col. 3 line 65 – Col. 4 line 4).

background sound level measurement means for measuring a level of an output from the background-sound microphone as a background-sound level (Col. 1 lines 21-29);

received-speech clarifying means for adjusting a gain for received speech (Fig. 2 item 207) that is output from the speaker based on the background-sound level measured by the background sound level measurement means, wherein the received speech that is output from the speaker is not received at a microphone of the speech communication apparatus (Col. 1 lines 29-41).

NOTE: For purposes of prior art, the microphone not collecting the output from the speaker is construed to be both functionally equivalent and equally effective as a speech communication system that does not implement feedback from a speaker to a microphone (Fig. 1 item 1). Urbanski does not teach any methods of feedback from final output to input microphone.

However, Urbanski fails to teach a transmission-speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal;

a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter

background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone

Suzuki teaches a circuit for compensating for frequency characteristic of microphone output which is arranged to have a combination of a microphone member of the pressure gradient type having a proximity effect represented by a rise in its low frequency range sensitivity as the microphone member approaches closer to a source of sound, and another microphone member of the pressure type developing no proximity effect, to cancel out the occurrence of proximity effect. A change in sensitivity of the pressure gradient type microphone in the low frequency range due to the proximity effect is determined from the level difference between the outputs of the two microphones. A signal representative of the difference is subtracted from the output of the pressure gradient type microphone thereby to effect a compensation for the proximity effect in the output of the pressure gradient type microphone (Suzuki Abstract & fig. 3).

Further, Suzuki teaches well known uses of a filter used to reduce the proximity effect by applying a frequency scale based on the proximity effect onto a signal (Suzuki Fig. 2,4, and 6), wherein the difference between the output of a microphone and the output of a filter are generated with respect to the proximity effect (Suzuki Fig. 1).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Urbanski to incorporate a transmission-

speech signal generation filter for manipulating a frequency characteristic of an output of the microphone to minimize a proximity effect produced in the output of the microphone, where the resulting signal output from the transmission- speech signal generation filter is a transmission-speech signal, a pseudo-proximity-effect filter for applying a pseudo proximity effect on the transmission-speech signal output by the transmission-speech signal generation filter, and background sound level measurement means for measuring a power level of background sound by subtracting the power of the output of the pseudo-proximity-effect filter from the power of the output of the microphone as taught by Suzuki to allow for noise reduction within a greater frequency range, wherein frequency response is independent of the minimization of the proximity effect, and can therefore reduce noise within the whole frequency range (Suzuki Col. 3 lines 38-52 & Fig. 3) whereby the proximity effect is compensated for through a subtractive filter means (Suzuki Fig. 1).

However, Urbanski in view of Suzuki fails to teach a background-sound microphone disposed at the rear face of the handset at almost the same height as the speaker, for collecting background sound (Todter Col. 8 lines 50-65 & Col. 9 lines 12-25);

Todter teaches a plurality of pick ups or microphones are provided (e.g. in a telephone handset) at known positions in relation to the source of the desired audio signal, so that all microphones will receive the desired audio signal (albeit possibly with gain and phase differences) and all microphones will receive the extraneous noise

signal (again possibly with gain and phase differences). The signals from each microphone may then be separately processed to provide electrical noise cancellation and added (possibly with appropriate weighting) to give the noise cancelled audio signal. The noise cancelling processing may make use of known propagation characteristic differences between the ambient noise field and the desired audio signal acoustic field.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Urbanski in view of Suzuki to as taught by Todter to allow for a user of the headset to be able to communicate and transmit data in response to incoming data, wherein using a microphone one or more microphones a headset would allow for the acquisition of the same noise signal but with various gains and phase differences, where several noise cancellation operations can be performed and summed to produce a desirable speech signal (Todter Col. 8 lines 50-65 & Col. 9 lines 12-25).

Re claim 11, Urbanski teaches the speech communication apparatus of claim 4, wherein the microphone is a unidirectional or bi-directional microphone (Fig. 1 item 101).

NOTE: The term microphone disclosed in general, is encompassing of any microphone polar pattern including but not limited to unidirectional or bidirectional microphone. Therefore, it would be necessary to utilize a microphone with a particular polar pattern as a matter of choice if and when directional preference is desirable.

Urbanski teaches a general microphone such as a unidirectional or bidirectional microphone on a mobile cellular phone (Col. 1 lines 42-56), where a bidirectional microphone could consist of two unidirectional microphones facing opposite one another.

6. Claims 14 and 15 are rejected under 35 U.S.C. 103(a) as being unpatentable over Urbanski, US 5,544,250 A (hereinafter Urbanski) in view of Soli et al. US 6563931 B1 (hereinafter Soli) and further in view of Todter et al US 5937070 A (hereinafter Todter).

Re claims 14, Urbanski teaches a speech communication apparatus for bi-directional speech communications, comprising:

a speaker for outputting received speech (Col. 1 lines 42-56)

a microphone for collecting speech to be transmitted (Fig. 1 item 101);

background sound level measurement calculator operable to measure a level of background sound (Col. 1 lines 21-29);

a received-speech clarifying section operable to adjust a gain for the received speech to be outputted by the speaker based on the level of the background sound measured by the background sound level measurement calculator (Col. 1 lines 29-41), wherein the received speech to be outputted by the speaker is not received by a microphone of the speech communication apparatus (Col. 3 line 65 – Col. 4 line 4)..

NOTE: For purposes of prior art, the microphone not collecting the output from the speaker is construed to be both functionally equivalent and equally effective as a

speech communication system that does not implement feedback from a speaker to a microphone (Fig. 1 item 1). Urbanski does not teach any methods of feedback from final output to input microphone. Additionally, for purposes of prior art, the proximity effect is construed to be both functionally equivalent and equally effective as noise produced from low frequency components picked up by the microphone in a changing environment, where the response of a system would be handled by an adaptive frequency response such as the adjustment of gain and reduction of noise.

However, Urbanski in view of Suzuki fails to teach a delay section operable to delay an output of a first background-sound microphone by a period of time corresponding to a delay time between transmission speech mixed into the output of the first background-sound microphone and transmission speech mixed into an output of a second background-sound microphone (Todter Col. 15 lines 1-15 & Fig. 12 item 72),

an adaptive filter (Todter Col. 13 lines 31-49) operable to estimate transmission of speech mixed into the output of the delay section (Todter Col. 15 lines 1-15 & Fig. 12 item 72),

an adder operable to subtract the transmission speech estimated by the adaptive filter from an output of the delay section (Todter Col. 15 lines 1-15 & Fig. 12 items 72 and 73),

a background sound level calculation section operable to calculate a level of an output of the adder and for outputting the result as the level of the background sound

(Todter Col. 15 lines 1-15 & Fig. 12 items 74 and summation node prior to output just below item 74).

Todter teaches a high pass frequency segment that provides for a weighted and phase-shifted sum of the "n" microphone signals; containing both phase correlated speaker's voice and uncorrelated external noise signals. The invention allows for the adjustment of signal weighting and phase shifting to amplify the speakers voice signal and attenuate the external noise. (59) The high frequency sections comprises a high pass filter 80 and a plurality of gain and delay blocks 81, 82, respectively, as well as a plurality of positive summing circuits 83. The low pass frequency filter may also comprise a plurality of circuits 71, 72, 73. (60) The low pass frequency segment of the invention provides for a weighted and phase-shifted subtraction of noise from the mouthpiece microphone signal. The invention allows for adjustment of gain weighting and phase shift to find the optimum improvement in signal to noise ratio, in any specific reverberant noise environment. (61) The block 60 further provides for the weighted summing via summing circuit 90 of low passed and high passed signals to reconstitute the total enhanced signal.

Additionally, Todter teaches an adaptive control block 9 that compares the noise cancellation signal output from the summing circuit 13 with the microphone signal to detect residual noise, and controls the gain compensation 20 and phase compensation 21 to keep the residual noise to a minimum. Also by negatively summing the audio signal into the adaptive control loop the loop will affect the compensation of the

cancellation loop block to provide a high fidelity output signal, notwithstanding quality of components, as discussed in the preamble.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a delay section utilizing several microphones where delayed speech from several microphones are mixed together, using an adaptive filter in for the subtraction of speech from a delay section. Delaying segments of speech would be necessary to add high pass filtered and low pass filtered components, where low pass components can be weighted and summed in addition to high pass components, to reduce the amount of noise in a signal. In a phase shifting environment, summation of delayed components from several microphones would allow for reduced noise, adjustable gain, and optimal signal to noise ratio conditions in a noisy or non-noisy environment.

Re claim 15, Urbanski in view of Soli fails to teach the speech communication apparatus of [[to]] claim 14, wherein the adaptive filter (Todter Col. 13 lines 31-49) estimates the transmission speech based on a difference between the output of the delay section and the transmission speech estimated by the adaptive filter (Todter Col. 15 lines 1-15 & Fig. 12 items 72 and 73).

Todter teaches a high pass frequency segment that provides for a weighted and phase-shifted sum of the "n" microphone signals; containing both phase correlated speaker's voice and uncorrelated external noise signals. The invention allows for the adjustment of signal weighting and phase shifting to amplify the speakers voice signal

and attenuate the external noise. (59) The high frequency sections comprises a high pass filter 80 and a plurality of gain and delay blocks 81, 82, respectively, as well as a plurality of positive summing circuits 83. The low pass frequency filter may also comprise a plurality of circuits 71, 72, 73. (60) The low pass frequency segment of the invention provides for a weighted and phase-shifted subtraction of noise from the mouthpiece microphone signal. The invention allows for adjustment of gain weighting and phase shift to find the optimum improvement in signal to noise ratio, in any specific reverberant noise environment. (61) The block 60 further provides for the weighted summing via summing circuit 90 of low passed and high passed signals to reconstitute the total enhanced signal.

Additionally, Todter teaches an adaptive control block 9 that compares the noise cancellation signal output from the summing circuit 13 with the microphone signal to detect residual noise, and controls the gain compensation 20 and phase compensation 21 to keep the residual noise to a minimum. Also by negatively summing the audio signal into the adaptive control loop the loop will affect the compensation of the cancellation loop block to provide a high fidelity output signal, notwithstanding quality of components, as discussed in the preamble.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention a delay section utilizing several microphones where delayed speech from several microphones are mixed together, using an adaptive filter in for the subtraction of speech from a delay section. Delaying segments of speech would be necessary to add high pass filtered and low pass filtered components, where low pass

components can be weighted and summed in addition to high pass components, to reduce the amount of noise in a signal. In a phase shifting environment, summation of delayed components from several microphones would allow for reduced noise, adjustable gain, and optimal signal to noise ratio conditions in a noisy or non-noisy environment.

Conclusion

7. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-

270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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